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EXAMINER	
MOORE, IAN N	
ART UNIT	**PAPER NUMBER**
2661	

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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)	
	09/778,186	HARDY, WILLIAM CHRISTOPHER	
	Examiner	Art Unit	
	Ian N Moore	2661	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 17 November 2004.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-32 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☒ Claim(s) 32 is/are allowed.

6)☒ Claim(s) 1-31 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on 17 November 2004 is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date 2-23-05.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Response to Amendment

1. The objections to the drawings (3A, 3B, 3C, and 9) are withdrawn since they are being amended accordingly.

2. Claim objections, on claim 21 and 32 are withdrawn since they are being amended accordingly.

3. Claim rejection under 35 USC § 112 second paragraph, on claims 12-14 are withdrawn since they are being amended accordingly.

4. Claims 1-31 are rejected by the same ground of rejections.

Claim Rejections - 35 USC § 102

5. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

6. Claims 1 is rejected under 35 U.S.C. 102(e) as being anticipated by Farris (U.S. 6,574,216).

Regarding claim 1, Farris'216 discloses a method for determining acceptability of quality of a second communications service (see FIG. 3, service quality provided by Internet 50 for terminals 90,11, 15), in comparison to a first

communications service (see FIG. 3, service quality provided by PSTN 10 for terminals 90,11, 15) which is deemed to exhibit acceptable quality, comprising the steps of

obtaining a first quality index (see col. 4, lines 63-67; a voice quality threshold/criteria of normal end-to-end voice circuit which a caller accepted) pertaining to the first communications service (see FIG. 3, PSTN 10, note the performance quality acceptable to the calling subscriber is determined based upon circuit switched performance quality threshold/criteria utilized in PSTN, thus, it is relevant to PSTN services; see col. 8, lines 15-40);

obtaining a second quality index (see col. 10, lines 1-24, 44-65; a measured threshold/criteria of the packet switched network; see col. 10, lines 1-24, 44-65), pertaining to the second communications service (see FIG. 3, Internet 50, note the performance quality threshold/criteria is determined in Internet, thus it is relevant to Internet services; see col. 10, lines 1-24); and

determining that the second communication service (see FIG. 3, routing the traffic over Internet 50) is of unacceptable quality if the second quality index differs from the first quality index service by more than a selected amount (see FIG. 6A, step 208 and 214, quality good? N; see col. 10, lines 1-2, 44 to col. 11, lines 15; note that the routing the call over the Internet is unacceptable when measured/tested quality threshold does not meet the PSTN acceptable quality threshold by a predetermined amount. Note that in order to perform a comparison and selecting

one service over the other, one must identify whether the differences are more than a predetermined amount).

Claim Rejections - 35 USC § 103

7. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

8. Claims 6 is rejected under 35 U.S.C. 103(a) as being unpatentable over Farris (U.S. 6,574,216) in view of Randic (U.S. 6,275,797).

Regarding claim 6, Farris'216 discloses a method for determining acceptability of quality of a second communications service (see FIG. 3, service quality provided by Internet 50 for terminals 90,11, 15), in comparison to a first communications service (see FIG. 3, service quality provided by PSTN 10 for terminals 90,11, 15) which is deemed to exhibit acceptable quality, comprising the steps of

obtaining a first quality index (see col. 4, lines 63-67; a voice quality threshold/criteria of normal end-to-end voice circuit which a caller accepted) pertaining to the first communications service (see FIG. 3, PSTN 10, note the performance quality acceptable to the calling subscriber is determined based upon circuit switched performance quality threshold/criteria utilized in PSTN, thus, it is relevant to PSTN services; see col. 8, lines 15-40);

determining the effect of at least one performance characteristic of the second communication service (see col. 10, lines 15-24, 45-60; the quality performance measures/characteristic such as flow rate and response time which has the effect on the voice-o-IP network are determined) upon a second quality index (see col. 10, lines 1-24, 44-65; a measured threshold/criteria of the packet switched network; see col. 10, lines 1-24, 44-65), pertaining to the second communications service (see FIG. 3, Internet 50, note the performance quality threshold/criteria is determined in Internet, thus it is relevant to Internet services; see col. 10, lines 1-24); and

determining that the second communication service (see FIG. 3, routing the traffic over Internet 50) is of unacceptable quality if the second quality index differs from the first quality index service by more than a selected amount (see FIG. 6A, step 208 and 214, quality good? N; see col. 10, lines 1-2, 44 to col. 11, lines 15; note that the routing the call over the Internet is unacceptable when measured/tested quality threshold does not meet the PSTN acceptable quality threshold by a predetermined amount. Note that in order to perform a comparison and selecting one service over the other, one must identify whether the differences are more than a predetermined amount).

Farris'216 does not explicitly disclose determining a value required to maintain the second quality index acceptable near the value of the first quality index.

However, the above-mentioned claimed limitations are taught by Randic'797. In particular, Randic'797 teaches obtaining a first quality index (see FIG. 1, voice test

file 23, which is the non-distorted common reference voice pattern, also see FIG. 3 steps 40 and 47; see col. 3, lines 15-39, see col. 6, lines 1-9),

obtaining a second quality index (see FIG. 1, transmitted voice test file 17, which is the distorted reference voice pattern, see FIG. 3, step 44) pertaining to the second communions service (see FIG. 1, WAN 11 or see FIG. 3, packet base network 42; test file 17 is routed through the packet switched network, thus it pertains/relevant to the packet switched network services; see col. 3, lines 40-60, see col. 5, lines 24-67; and

determining a value (see FIG. 3, step 52, determining voice quality factor 52, and voice quality factor 27; also see FIG. 1), required to maintain the second quality index acceptable near the value of the first quality index (see col. 7, lines 1-29; note that a voice quality factor 27 is determine to improve and conform the reference acceptable threshold level of voice path defined by voice test file 23 (i.e. 75% match to the reference)).

Note that Farris'216 teaches a quality test application device, which monitored and measured the VoIP signal quality parameters over Internet and compare the quality parameters between the PSTN acceptable quality parameter against measured Internet quality parameter. Randic'797 teaches utilizing the voice test signal to measure the voice quality parameter over the packet switched network and performing determination according to threshold. In view of this, having the system of Farris'216 and then given the teaching of Randic'797, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify

the system of Randic'797, for the purpose of providing testing/measuring mechanism of the voice quality over the Internet and determination a quality factor result in order to improve the quality of voice path to conform to the reference level, as taught by Randic'797, since Randic'797 states the advantages/benefits at see col. 1, lines 14-20, col. 2, lines 5-26, 29-55 that it would provide the voice quality factor which indicates transmission and processing quality of the communication link. The motivation being that by measuring and testing the voice communication paths over the Internet and utilizes the results to improve the quality of voice path, it can increase capability of the service provider to proactively test and measure the voice quality of the traffic sent over a packet network before the signal is totally degraded.

9. Claims 4 and 5 are rejected under 35 U.S.C. 103(a) as being unpatentable over Farris'216 and Randic'797, in view of Giers (U.S. 4,015,480).

Regarding claims 4 and 5, Farris'216 discloses measuring performance characteristics of the second network and computing an quality index (see FIG. 6A, step 204; 206; during quality check and monitor step, a measured performance quality level (i.e. measured rate or measured response time) is acquired by the test signal from quality test application 122, see FIG. 5, col. 10, lines 1-24, 44-65. Also, see FIG. 3, Internet 50, note the performance quality threshold/criteria is determined in Internet; see col. 10, lines 1-24).

Farris'216 does not explicitly discloses an expected quality index (see Randic'797 FIG. 3, step 52, determining voice quality factor 52, and voice quality

factor 27; also see FIG. 1) for the second communications service (see FIG. 3, Internet 50 service); see col. 7, lines 1-29.)

Note that Farris'216 teaches a quality test application device, which monitored and measured the VoIP signal quality parameters over Internet and compare the quality parameters between the PSTN acceptable quality parameter against measured Internet quality parameter. Randic'797 teaches utilizing the voice test signal to measure the voice quality parameter over the packet switched network and performing determination according to threshold. In view of this, having the system of Farris'216 and then given the teaching of Randic'797, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the system of Randic'797, for the purpose computing expected quality factor, as taught by Randic'797, since Randic'797 states the advantages/benefits at see col. 1, lines 14-20, col. 2, lines 5-26, 29-55 that it would provide testing and measuring voice path quality in a communication networks which generated reproducible, objective, and easily evaluated. The motivation being that by measuring and testing the voice communication paths over the Internet, it can increase capability of the service provider to proactively test and measure the voice quality of the traffic sent over a packet network before the signal is totally degraded.

Neither Farris'216 nor Randic'797 explicitly discloses applying an effects matrix.

However, the above-mentioned claimed limitations are taught by Giers'480. In particular, Giers'480 teaches computing result (see col. 2, lines 20-35, Abstract;

balanced result) by applying an effects matrix (see col. 2, lines 23-29; reference values) to the measured value (see col. 2, lines 21-39; measured unbalance signal; note that measured unbalance signals are multiplied by the reference values in order to result a balance signal).

Note that the combined system of Farris'216 and Randic'797 teaches the expected or resulted quality index factor. Gier'480 teaches balancing the result by multiplying with reference values and the measured values. In view of this, having the combined system of Farris'216 and Randic'797, then given the teaching of Giers'480, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216 and Randic'797, for the purpose of multiplying the measured signal with the reference signal, as taught by Giers'480, since Giers'480 states the advantages/benefits at col. 1, lines 50 to col. 2, lines 20 that it would provide a balancing which deliver the unbalance information with reliable high speed and precision . The motivation being that by multiplying the measured values with the reference values, it can increase the smooth estimation of expected quality factor by utilizing the balancing method.

10. Claim 9 and 10 are rejected under 35 U.S.C. 103(a) as being unpatentable over Farris'216 and Randic'797, as applied to claim 6 above, and further in view of Sand'746.

Regarding claims 9 and 10, the combined system of Farris'216 and Randic'797 discloses wherein said performance characteristics.

Neither Farris'216 nor Randic'797 explicitly discloses packet loss rate (see Sand'746 FIG. 4, network performance packet loss 4 and voice performance echo path loss 4) and packet delay (see Sand'746 FIG. 4, network performance packet delay 5 and voice performance echo path delay 3) see col. 6, lines 20-35.

However, the above-mentioned claimed limitations are taught by Sand'746. In view of this, having the system of Farris'216 and Randic'797, then given the teaching of Sand'746, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216 and Randic'797, for the purpose of providing testing/measuring mechanism of the voice quality such as packet loss and delay over the Internet and determination a result in order to improve the quality of voice path to conform to the SLA and ANSI level, as taught by Sand'746, since Sand'746 states the advantages/benefits at see col. 1, lines 46 col. 3, lines 67 that it would provide meaningful and accurate measurement of voice GOS which can impact the performance. The motivation being that by measuring and testing the voice communication paths over the Internet and utilizes the results to improve the quality of voice path, it can increase capability of the service provider to proactively test and measure the voice quality of the traffic sent over a packet network before the signal is totally degraded, and the measuring process can be provided through software.

11. Claims 11, 17 and 18 are rejected under 35 U.S.C. 103(a) as being unpatentable over Farris'216, Sand'746 in view of Oouchi'203.

Regarding claim 11, Farris'216 discloses a method for determining quality performance of a second communications service (see FIG. 3, service quality provided by Internet 50 for terminals 90,11, 15), in comparison to a first communications service (see FIG. 3, service quality provided by PSTN 10 for terminals 90,11, 15), comprising the steps of

obtaining a first quality index (see col. 4, lines 63-67; a voice quality threshold/criteria of normal end-to-end voice circuit which a caller accepted) pertaining to the first communications service (see FIG. 3, PSTN 10, note the performance quality acceptable to the calling subscriber is determined based upon circuit switched performance quality threshold/criteria utilized in PSTN, thus, it is relevant to PSTN services; see col. 8, lines 15-40);

determining the effect of at least one performance characteristic of the second communication service (see col. 10, lines 15-24, 45-60; the quality performance measures/characteristic such as flow rate and response time which has the effect on the voice-o-IP network are determined) upon a second quality index (see col. 10, lines 1-24, 44-65; a measured threshold/criteria of the packet switched network; see col. 10, lines 1-24, 44-65), pertaining to the second communications service (see FIG. 3, Internet 50, note the performance quality threshold/criteria is determined in Internet, thus it is relevant to Internet services; see col. 10, lines 1-24); and

determining that the performance characteristic of the second communication service (see FIG. 3, routing the traffic over Internet 50) is of unacceptable quality if

the second quality index differs from the first quality index service (see FIG. 6A, step 208 and 214, quality good? N; see col. 10, lines 1-2, 44 to col. 11, lines 15; note that the routing the call over the Internet is unacceptable when measured/tested quality level does not meet the prestored acceptable quality level).

Farris'216 does not explicitly disclose determining the effect of a first (Sand'746 see FIG. 4, determining/measuring network performance parameter 4, packet loss) and second performance characteristic (Sand'746 see FIG. 4, determining/measuring network performance parameter 5, packet delay at the IP telephony measurement device; see col. 5, lines 24-55) and

assuming a selected value for the performance characteristic (see FIG. 4, ANSI standard performance parameter is selected for network work performance table),

determining a value (see Sand'746 FIG. 4, step 56, a computed result parameter, see col. 5, lines 45-50) see col. 6, lines 20-40; a computed performance result for Internet telephony services) required to maintain the second quality index (see FIG. 4, the table/list that contains voice performance level of Internet telephony service, such as packet loss, delay and jitter and their corresponding scores) acceptable near the value of the first quality index (see FIG. 4, ANSI T.221 and ANSI T1 LB 566 GOS performance parameter required for PSTN service; see col. 6, lines 20-35; note that GOB (good or better) scores are determined by comparing ANSI T1 LB standard with the measured performance parameters in order to maintain SLA, service level agreements.)

However, the above-mentioned claimed limitations are taught by Sand'746. Note that Farris'216 teaches a quality test application device, which monitored and measured the VoIP signal quality parameters over Internet and compare the quality parameters between the PSTN acceptable quality parameter against measured Internet quality parameter. Sand'746 teaches utilizing IP telephony measurement device to measure the voice quality parameter over the Internet. In view of this, having the system of Farris'216 and then given the teaching of Sand'746, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the system of Farris'216, for the purpose of providing testing/measuring mechanism of the voice quality over the Internet and determination a result in order to improve the quality of voice path to conform to the SLA and ANSI level, as taught by Sand'746, since Sand'746 states the advantages/benefits at see col. 1, lines 46 col. 3, lines 67 that it would provide meaningful and accurate measurement of voice GOS which can impact the performance. The motivation being that by measuring and testing the voice communication paths over the Internet and utilizes the results to improve the quality of voice path, it can increase capability of the service provider to proactively test and measure the voice quality of the traffic sent over a packet network before the signal is totally degraded, and the measuring process can be provided through software.

Neither Farris'216 nor Sand'746 explicitly discloses in the context of the selected value for the first performance (see Oouchi'203 col. 2, lines 40-44; first threshold parameter), determining a value for the second performance (see

Oouchi'203 col. 2, lines 48-50; a second threshold larger than the above first threshold; see Oouchi'203 col. 2, lines 34 to col. 3, lines 17; note that the second threshold is determined according the first threshold in order to maintain and the control the rate within the threshold level).

However, the above-mentioned claimed limitations are taught by Oouchi'203. In view of this, having the combined system of Farris'216 and Sand'746, then given the teaching of Oouchi'203, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216 and Sand'746, for the purpose of providing defining the second threshold based upon the first threshold in order to maintain the quality of service, as taught by Oouchi'203, since Oouchi'203 states the advantages/benefits at col. 2, lines 25-34 that it would suppress congestion caused by the violation cells in a packet network and assure of quality of service. The motivation being that by detecting utilizing threshold in order to maintain the quality of service, it can increase the network throughput and utilization.

Regarding claims 17 and 18, Sand'746 discloses packet loss rate (see Sand'746 FIG. 4, network performance packet loss 4 and voice performance echo path loss 4) and packet delay (see Sand'746 FIG. 4, network performance packet delay 5 and voice performance echo path delay 3) see col. 6, lines 20-35.

In view of this, having the system of Farris'216 and then given the teaching of Sand'746, it would have been obvious to one having ordinary skill in the art at the

time the invention was made to modify the system of Farris'216 as taught by Sand'746, for the same purpose and motivation as described above in claim 11.

12. Claims 12-14 are rejected under 35 U.S.C. 103(a) as being unpatentable over Farris'216, Sand'746 and Oouchi'203, as applied to claim 11 above, further in view of well established teaching in art.

Regarding claims 12-14, Sand'746 teaches wherein said first performance characteristic (Sand'746 see FIG. 4, determining/measuring network performance parameter 5, packet delay at the IP telephony measurement device; see col. 5, lines 24-55) and second performance characteristic (Sand'746 see FIG. 4, determining/measuring network performance parameter 4, packet loss).

Neither Farris'216, Sand'746 nor Oouchi'203 explicitly discloses selecting a first performance characteristic which has an effect upon the second quality index that is substantially independent of the second performance characteristic or any other performance characteristics.

However, the above-mentioned claimed limitations are taught by well-established teaching in art. Well-established teaching in art teaches selecting a first performance characteristic which has an effect upon the second quality index that is substantially independent of the second performance characteristic or any other performance characteristics. Sand'746 teaches the packet delay (i.e. first performance characteristic) and packet lost (i.e. second performance characteristic). It is well known in the art that the packet delay has an effect on the network

performance and quality in the voice-over-IP network. Also, it is well known in the art that the packet delay is independently measured, and it is different or independent from any other quality parameters (see Sand'746 FIG. 4, network performance jitter 6 or packet loss 4) or the packet loss (see Sand'746 FIG. 4, packet loss 4).

In view of this, having the combined system of Farris'216, Sand'746 and Oouchi'203, then given the teaching of well established teaching in art, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216, Sand'746 and Oouchi'203, for the purpose of providing the mechanism that the packet delay has effect on the quality of the network, and the packet delay is different from any other network performance parameters, as taught by well established teaching in art. The motivation being that by the detection and determining the packet delay, it can increase the network testing capability by prioritizing the potential failures and enhance the determination of the required minimum acceptable quality of service.

13. Claim 19 is rejected under 35 U.S.C. 103(a) as being unpatentable over Farris (U.S. 6,574,216) and Randic (U.S. 6,275,797), in view of well established teaching in art.

Regarding claim 19, Farris'216 discloses a method for determining quality performance required of a second communications service (see FIG. 3, service quality provided by Internet 50 for terminals 90,11, 15), in comparison to a first

communications service (see FIG. 3, service quality provided by PSTN 10 for terminals 90,11, 15), comprising the steps of

obtaining a first quality index (see col. 4, lines 63-67; a voice quality threshold/criteria of normal end-to-end voice circuit which a caller accepted) pertaining to the first communications service (see FIG. 3, PSTN 10, note the performance quality acceptable to the calling subscriber is determined based upon circuit switched performance quality threshold/criteria utilized in PSTN, thus, it is relevant to PSTN services; see col. 8, lines 15-40);

obtaining a second quality index representing the quality (see col. 10, lines 1-24, 44-65; a measured threshold/criteria of the packet switched network; see col. 10, lines 1-24, 44-65) of the second communications service (see FIG. 3, Internet 50, note the performance quality is determined in Internet, thus it is relevant to Internet services; see col. 10, lines 1-24) subjected to at least one performance characteristic of the second communication service (see col. 10, lines 15-24, 45-60; the quality performance measures/characteristic such as flow rate and response time which has the effect on the voice-o-IP network are determined),

determining that the second communication service (see FIG. 3, routing the traffic over Internet 50) is of unacceptable quality if the second quality index differs from the first quality index service (see FIG. 6A, step 208 and 214, quality good? N; see col. 10, lines 1-2, 44 to col. 11, lines 15; note that the routing the call over the Internet is unacceptable when measured/tested quality level does not meet the prestored acceptable quality level).

Farris'216 does not explicitly disclose first communications service occurring without performance characteristic (see Randic'797 FIG. 1, voice test file 23 utilized as non-distorted common voice reference pattern from the circuit switched service, also see Randic'797FIG. 3 steps 40 and 47; see col. 3, lines 15-39, see col. 6, lines 1-9), and second communications service occurring with the performance characteristic subjected to at least on degraded performance characteristic (see Randic'797 FIG. 1, transmitted voice test file 17, which is the distorted voice pattern, see Randic'797 FIG. 3, step 44; see FIG. 1, WAN 11 or see FIG. 3, packet base network 42; test file 17 is routed through the packet switched network, thus it pertains/relevant to the packet switched network services; see Randic'797 col. 3, lines 40-60, see col. 5, lines 24-67); and

determining an quality index for communication occurring through the second communication network (see Randic'797 FIG. 3, step 52, determining voice quality factor 52 and voice quality factor 27, see FIG. 1; note that a quality factor required for the packet base network is determined),

said quality index being an value resulting from a mixture of first communication occurring without the degraded performance characteristic and second communication occurring with the degraded performance characteristic (see Randic'797 FIG. 3, step 48 and 52; a voice quality factor is determined based upon the comparison of mixed/both distorted voice pattern from the packet based network and common non-distorted reference voice pattern (which normally utilized in circuit switched network); see Randic'797 col. 6, lines 10-36).

expressing the required quality performance of the second communication service as a proportion (see Randic'797 col. 7, lines 10-21; see col. 6, lines 30-43; the quality factor 75% which is 3 out 4 matching words) between said first communications and said second communications required to maintain said quality index acceptably near the value of the first quality index (see Randic'797 col. 7, lines 1-29; note that a voice quality factor 27 is determined to improve and conform the reference acceptable threshold level of voice path defined by voice test file 23 (i.e. 75% match to the reference)).

However, the above-mentioned claimed limitations are taught by Randic'797. Note that Farris'216 teaches a quality test application device, which monitored and measured the VoIP signal quality parameters over Internet and compare the quality parameters between the PSTN acceptable quality parameter against measured Internet quality parameter. Randic'797 teaches utilizing the voice test signal to measure the voice quality parameter over the packet switched network and performing determination according to threshold. In view of this, having the system of Farris'216 and then given the teaching of Randic'797, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the system of Randic'797, for the purpose of providing testing/measuring mechanism of the voice quality over the Internet and determination a quality factor result in order to improve the quality of voice path to conform to the reference level, as taught by Randic'797, since Randic'797 states the advantages/benefits at see col. 1, lines 14-20, col. 2, lines 5-26, 29-55 that it would provide the voice quality factor which

indicates transmission and processing quality of the communication link. The motivation being that by measuring and testing the voice communication paths over the Internet and utilizes the results to improve the quality of voice path, it can increase capability of the service provider to proactively test and measure the voice quality of the traffic sent over a packet network before the signal is totally degraded.

Neither Farris'216 nor Sand'746 explicitly discloses determining an averaged composite quality index for communication occurring through the second communication, said average composite quality index being an value resulting from a mixture of first communication and second communication, expressing the required quality performance of the second communication service to maintain said average composite quality index.

However, the above-mentioned claimed limitations are taught by well-established teaching in art. In particular, well established teaching in art teaches determining an averaged composite quality index for communication occurring through the second communication and said average composite quality index being an value resulting from a mixture of first communication and second communication. Note that Randic'797 teaches determining the quality of voice signals by utilizing statistical interpretation of human listeners called mean opinion scores (MOS), which are being utilized by the computers in see col. 2, lines 10-23. Randic'797 further teaches determining a quality factor by utilizing the distorted signal from the packet based network and the non-distorted signal commonly used in circuit switch network. It is well known in the art, when by taking average between two parameters (i.e.

distorted and non-distorted patterns) in order to determine a value that is commonly acceptable to both parameters by avoiding the extremes the high and low margins/threshold. In order to determine a minimum acceptable threshold in the packet based network, one can utilized readily available mathematical process "averaging" (also taught by Randic'797 as means) between the distorted and non-distorted patterns. Well established teaching in art teaches expressing the required quality performance of the second communication service to maintain said average composite quality index acceptably near the value of the first quality index. The average threshold pattern for the packet based network must be closed to circuit switched network since the non-distorted common reference pattern is utilized during averaging.

In view of this, having the combined system of Farris'216 and Sand'746, then given the teaching of well established teaching in art, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216 and Sand'746, for the purpose of providing averaging between two parameter on order to determine an acceptable value which is common and close to both parameters, as taught by well established teaching in art. The motivation being that by taking an average between two parameters when determining the minimum threshold, it can avoid the extremes high and low margins/threshold in voice pattern used in the packet based network, thereby, ensuring the acceptable voice quality for subscriber which increase the customer satisfaction.

14. Claim 21 is rejected under 35 U.S.C. 103(a) as being unpatentable over Farris (U.S. 6,574,216) and Randic (U.S. 6,275,797), in view of ITU P.830.

Regarding claim 21, the combined system of Farris'216, Randic'797 and well established teaching in art first, second and averaged composite quality indices and the percentage of total acceptable quality (i.e. P values) as described above in claim 19.

Neither Farris'216 nor Randic'797 explicitly discloses wherein quality indices are values assoicatd with a percentage (P) of calls or connection that elicit unusable (U), difficult (D), or irritating (I) responses P(UDI) (see ITU P.830, see page 14-15, section 10.2, 10.2.1, listening quality scales or rating: wherein Unusable is bad, Difficult is poor, and Irritating is fair).

However, the above-mentioned claimed limitations are taught by ITU P.830. In view of this, having the combined system of Farris'216 and Randic'797, then given the teaching of ITU P.830, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216 and Randic'797, for the purpose of providing voice quality scales/rating, as taught by ITU P.830, since ITU P.830 states the advantages/benefits at see page ii, introduction, paragraph 2 that it would ensure the performance of the complete system satisfactory. The motivation being that by testing and measuring the degradation contributed by the non-linear part of the transmission path, it can

increase the reliability of the network and ability to estimate telephone connection performances.

15. Claim 22 is rejected under 35 U.S.C. 103(a) as being unpatentable over Sand'746 and Shaffer (U.S. 5,898,668), in view of well established teaching in art.

Regarding claim 22, Sand'746 discloses a method for determining how a first performance characteristic (see FIG. 4, network performance packet delay 5 and voice performance echo path delay 3) having a given value affects the quality of a communication service (see FIG. 4, steps 44, 52 and 54; see FIG. 4, note that the standard delay values defined in ANSI T.221 and ANSI T1 LB 566 GOS standard are determined according to the acceptable QoS in industry; see col. 6, lines 20-35), the method comprising:

obtaining an original data set pertaining to occurrences of various values of at least one second performance characteristic within the communication service (see FIG. 4, steps 44, 52 and 54; see FIG. 4, note that the standard/original ANSI T.221 and ANSI T1 LB 566 GOS contains the network performance packet loss 4 and echo path loss 4 values; see col. 6, lines 20-35) and

assuming the first performance characteristic is set to said given value (see FIG. 4, steps 44, 52 and 54; see FIG. 4, note that the standard delay values defined in ANSI T.221 and ANSI T1 LB 566 GOS standard are determined according to the acceptable QoS in industry; see col. 6, lines 20-35).

computing a quality index for the communication service (see Sand'746 FIG. 4, step 56, a computed result parameter, see col. 5, lines 45-50) see col. 6, lines 20-40; a computed performance result for Internet telephony services) based upon the data set (see FIG. 4, steps 54 and 56; see col. 6, lines 20-35; note that GOB (good or better) scores are determined based upon ANSI T1 LB standard and the measured performance parameters in order to maintain SLA, service level agreements.)

Sand'746 does not explicitly disclose computing an altered data set by changing (see Shaffer'668 FIG. 3, step 80; present QoS Table is updated), in the original data set (see Shaffer'668 FIG. 2, Present time QoS table), the occurrences of values of the second performance characteristic (see Shaffer'668 col. 6, lines 5-44; delay, latency, loss; note that based upon delay arrival or lost packet is assessed, the present time QoS table is updated).

However, the above-mentioned claimed limitations are taught by Shaffer'668. In view of this, having the system of Sand'746 and then given the teaching of Shaffer'668, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the system of Sand'746, for the purpose of updating the present quality of service table according to packet lost or delay, as taught by Shaffer'668, since Shaffer'668 states the advantages/benefits at see col. 3, lines 15-67 that it would maximize the QoS at a tariff that is acceptable to users. The motivat7ion being that by updating present QoS table based upon monitored

quality parameter such as packet lost or delay, it can increase monitoring capability by having up-to-data status of the network.

Neither Sand'746 nor Shaffer'668 explicitly discloses determining the effect that the first performance characteristic has upon the occurrences of values of the second performance characteristic.

However, the above-mentioned claimed limitations are taught by well-established teaching in art. In particular, well-established teaching in art teaches determining the effect that the first performance characteristic has upon the occurrences of values of the second performance characteristic. Sand'746 teaches the packet delay (i.e. first performance characteristic) and packet lost (i.e. second performance characteristic). Shaffer'668 teaches updated the present/original table based upon occurrence of packet delay or lost. It is well known in the art that the number of packet delay has effect upon the number of packet lost. When the packet delay is encountered and detected, subsequently the packet lost will be occurred and detected.

In view of this, having the combined system of Sand'746 and Shaffer'668, then given the teaching of well established teaching in art, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Sand'746 and Shaffer'668, for the purpose of providing the mechanism that the effect that the packet delay has upon the occurrence of packet lost, as taught by well established teaching in art. The motivation being that by the detection and determining the packet delay and

subsequently the packet loss, it can increase the network testing capability by prioritizing the potential failures and enhance the determination of the required minimum acceptable quality of service.

16. Claims 23 is rejected under 35 U.S.C. 103(a) as being unpatentable over Sand'746 and Shaffer'668, as applied to claims 22 above, and further in view of Giers (U.S. 4,015,480).

Regarding claim 23, the combined system of Sand'746 and Shaffer'668 discloses computing a quality index is performed on the altered data set as described above in claim 22.

Neither Sand'746 nor Shaffer'668 explicitly discloses applying an effects matrix.

However, the above-mentioned claimed limitations are taught by Giers'480. In particular, Giers'480 teaches computing result (see col. 2, lines 20-35, Abstract; balanced result) by applying an effects matrix (see col. 2, lines 23-29; reference values) to the measured value (see col. 2, lines 21-39; measured unbalance signal; note that measured unbalance signals are multiplied by the reference values in order to result a balance signal).

Note that the combined system of Sand'746 and Shaffer'668 teaches the resulted altered data set. Gier'480 teaches balancing the result by multiplying with reference values and the measured values. In view of this, having the combined system of Sand'746 and Shaffer'668, then given the teaching of Giers'480, it would

have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Sand'746 and Shaffer'668, for the purpose of multiplying the measured signal with the reference signal, as taught by Giers'480, since Giers'480 states the advantages/benefits at col. 1, lines 50 to col. 2, lines 20 that it would provide a balancing which deliver the unbalance information with reliable high speed and precision . The motivation being that by multiplying the measured values with the reference values, it can increase the smooth estimation of expected quality factor by utilizing the balancing method.

17. Claims 2, 3,7,8,20,26 and 27 are rejected under 35 U.S.C. 103(a) as being unpatentable over Farris'216 and Randic'797, as applied to claims 1, 6, or 19 above, and further in view of ITU-T P.830.

Regarding claims 2, 7, and 20, Randic'797 discloses means opinion scores (see col. 2, lines 10-26; mean opinion scores, MOS; note that MOS is traditionally performed by human listeners for end-to-end testing. Randic'797 disclose the test apparatus of sending and receiving voice test files, instead of human listeners.

Neither Farris'216 nor Randic'797 explicitly disclose wherein quality indices are mean opinion scores (see ITU P.830, see page 5, section 8, paragraph 1, Means Opinion Scores (MOS) and see page 14-15, section 10.2, opinion scales and tables utilizes in testing where MOS is determined).

However, the above-mentioned claimed limitations are taught by ITU P.830. In view of this, having the combined system of Farris'216 and Randic'797, then given

the teaching of ITU P.830, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216 and Randic'797, for the purpose of providing MOS to measure means opinions, as taught by ITU P.830, since ITU P.830 states the advantages/benefits at see page ii, introduction, paragraph 2 that it would ensure the performance of the complete system satisfactory. The motivation being that by testing and measuring the degradation contributed by the non-linear part of the transmission path, it can increase the reliability of the network and ability to estimate telephone connection performances.

Regarding claims 26, Farris'216 discloses a method for determining acceptability of quality of a second communications service (see FIG. 3, service quality provided by Internet 50 for terminals 90,11, 15), in comparison to a first communications service (see FIG. 3, service quality provided by PSTN 10 for terminals 90,11, 15), which exhibit acceptable quality, comprising the steps of

measuring at least one performance characteristic for the first communication service (see col. 4, lines 63-67; a voice quality threshold/criteria of normal end-to-end voice circuit which a caller accepted) is determined. Also during quality check step, a predefined performance quality level (i.e. acceptable rate or acceptable response time) according to CPR of calling subscriber is utilized by quality test application 122, see FIG. 5; see col. 10, lines 1-4, see col. 10, lines 59-62) pertaining to the first communications service (see FIG. 3, PSTN 10, note the performance quality acceptable to the calling subscriber is determined based upon

circuit switched performance quality utilized in PSTN, thus, it is relevant to PSTN services; see col. 8, lines 15-40);

measuring at least one performance characteristic for the second communication service (see col. 10, lines 1-24, 44-65; a measured threshold/criteria of the packet switched network; see col. 10, lines 1-24, 44-65, Note that during quality check and monitor step, a measured performance quality level (i.e. measured rate or measured response time) is acquired by the test signal from quality test application 122, see FIG. 5, col. 10, lines 1-24, 44-65), pertaining to the second communications service (see FIG. 3, Internet 50, note the performance quality is determined in Internet, thus it is relevant to Internet services; see col. 10, lines 1-24); and

determining that the second communication service (see FIG. 3, routing the traffic over Internet 50) is of unacceptable quality if the second quality index is more than a perceptible difference threshold from the first quality index (see FIG. 6A, step 208 and 214, quality good? N; see col. 10, lines 1-2, 44 to col. 11, lines 15; note that the routing the call over the Internet is unacceptable when measured/tested quality threshold does not meet the PSTN acceptable quality threshold by a predetermined amount. Note that in order to perform a comparison and selecting one service over the other, one must identify whether the difference is more than a predetermined amount).

Farris'216 does not explicitly disclose second score/result (see Randic'797 FIG. 1, transmitted voice test file 17) is less than the first score/result (see

Randic'797 FIG. 1, voice test file 23; computer 14 compares the transmitted voice test file 17 and the reference voice test file 23; see Randic'797 col. 7, lines 1-29; note that no communication is initiated via Internet when the compared result quality factor 27 is less than threshold factor of 75%. Note that when the voice quality is lesser than the acceptable threshold quality factor value, it means the noise/packet loss is more than a threshold value which it can tolerate/accept).

However, the above-mentioned claimed limitations are taught by Randic'797. Note that Farris'216 teaches a quality test application device, which monitored and measured the VoIP signal quality parameters over Internet and compare the quality parameters between the PSTN acceptable quality parameter against measured Internet quality parameter. Randic'797 teaches utilizing the voice test signal to measure the voice quality parameter over the packet switched network and performing determination according to threshold. In view of this, having the system of Farris'216 and then given the teaching of Randic'797, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the system of Randic'797, for the purpose of providing testing/measuring mechanism of the voice quality over the Internet and determination whether the quality of transmitted voice test file is less than the quality of reference voice file, as taught by Randic'797, since Randic'797 states the advantages/benefits at see col. 1, lines 14-20, col. 2, lines 5-26, 29-55 that it would provide the voice quality factor which indicates transmission and processing quality of the communication link. The motivation being that by measuring and testing the voice communication paths over

the Internet and utilizes the results to improve the quality of voice path, it can increase capability of the service provider to proactively test and measure the voice quality of the traffic sent over a packet network before the signal is totally degraded.

Neither Farris'216 nor Randic'797 explicitly disclose wherein quality indices are mean opinion scores (see ITU P.830, see page 5, section 8, paragraph 1, Means Opinion Scores (MOS) and see page 14-15, section 10.2, opinion scales and tables utilizes in testing where MOS is determined).

Note that Farris'216 teaches comparing and determining Internet measured Internet quality value/threshold is more than PSTN acceptable quality threshold by a predefined amount. Randic'797 teaches determining if Internet quality score/result is less than reference quality score/result. However, the above-mentioned claimed limitations are taught by ITU P.830. ITU P.830 teaches determining MOS with weighted factors to determine the quality. Thus, the combined system of Farris'216 and Randic'797 quality determination by utilizing MOS with respect to difference threshold. In view of this, having the combined system of Farris'216 and Randic'797, then given the teaching of ITU P.830, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216 and Randic'797, for the purpose of providing MOS to measure means opinions, as taught by ITU P.830, since ITU P.830 states the advantages/benefits at see page ii, introduction, paragraph 2 that it would ensure the performance of the complete system satisfactory. The motivation being that by testing and measuring the degradation contributed by the non-linear part of the

transmission path, it can increase the reliability of the network and ability to estimate telephone connection performances.

Regarding claims 3, 8 and 27, Randic'797 discloses wherein said first and second quality indices relate to an average proportion of communications that would be rated as acceptable by users (see col. 2, lines 10-26; mean opinion scores, MOS; note that MOS is traditionally performed by human listeners for end-to-end testing. The average/mean percentage/proportion of communication that would be acceptable by user is determined as 75% of match words; see col. 6, lines 29-35).

Neither Farris'216 nor Randic'797 explicitly disclose objectionable by users (see ITU P.830, see page 5, section 8, paragraph 1, Means Opinion Scores (MOS) and see page 14-15, section 10.2, opinion scales and tables utilizes the opinion of users (i.e. listeners and talkers) regarding the quality of speech as poor (opinion score 2) and bad (opinion score 1), and both poor and bad qualities are not acceptable to the user and they can be rated as objectionable by users; see page 6, section 8.1.3 talkers and page 10, sections 10.1 and 11).

However, the above-mentioned claimed limitations are taught by ITU P.830. In view of this, having the combined system of Farris'216 and Randic'797, then given the teaching of ITU P.830, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216 and Randic'797, for the purpose of providing the average score rating of poor and bad opinions, as taught by ITU P.830, since ITU P.830 states the advantages/benefits at see page ii, introduction, paragraph 2 that it would ensure the

performance of the complete system satisfactory. The motivation being that by testing and measuring the degradation contributed by the non-linear part of the transmission path, it can increase the reliability of the network and ability to estimate telephone connection performances.

18. Claims 15,16, 24 and 25 are rejected under 35 U.S.C. 103(a) as being unpatentable over Farris'216 and Sand'746, as applied to claims 11 or 22 above, and further in view of ITU-T P.830.

Regarding claims 16 and 24, Sand'746 discloses opinion scores (see col. 1, lines 40-45; customer opinion performance in percentage, which is also known as Grade of service, GOS. Sand'746 disclose the test apparatus of sending and receiving voice over the Internet.

Neither Farris'216 nor Sand'746 explicitly disclose wherein quality indices are mean opinion scores (see ITU P.830, see page 5, section 8, paragraph 1, Means Opinion Scores (MOS) and see page 14-15, section 10.2, opinion scales and tables utilizes in testing where MOS is determined).

However, the above-mentioned claimed limitations are taught by ITU P.830. In view of this, having the combined system of Farris'216 and Sand'746, then given the teaching of ITU P.830, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216 and Sand'746, for the purpose of providing MOS to measure means opinions, as taught by ITU P.830, since ITU P.830 states the advantages/benefits at see page ii, introduction, paragraph 2 that it would ensure the performance of the

complete system satisfactory. The motivation being that by testing and measuring the degradation contributed by the non-linear part of the transmission path, it can increase the reliability of the network and ability to estimate telephone connection performances.

Regarding claims 15 and 25, Sand'746 discloses wherein said first and second quality indices relate to an proportion of communications that would be rated as acceptable by users (see col. 1, lines 40-45; customer opinion performance in percentage, which is also known as Grade of service, GOS. The percentage/portion of good or better (%GOB) scores, which are acceptable to user, related to voice GOS is calculated).

Neither Farris'216 nor Sand'746 explicitly disclose objectionable by users (see ITU P.830, see page 5, section 8, paragraph 1, Means Opinion Scores (MOS) and see page 14-15, section 10.2, opinion scales and tables utilizes the opinion of users (i.e. listeners and talkers) regarding the quality of speech as poor (opinion score 2) and bad (opinion score 1), and both poor and bad qualities are not acceptable to the user and they can be rated as objectionable by users; see page 6, section 8.1.3 talkers and page 10, sections 10.1 and 11).

However, the above-mentioned claimed limitations are taught by ITU P.830. In view of this, having the combined system of Farris'216 and Sand'746, then given the teaching of ITU P.830, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216 and Sand'746, for the purpose of providing the average score rating of

poor and bad opinions, as taught by ITU P.830, since ITU P.830 states the advantages/benefits at see page ii, introduction, paragraph 2 that it would ensure the performance of the complete system satisfactory. The motivation being that by testing and measuring the degradation contributed by the non-linear part of the transmission path, it can increase the reliability of the network and ability to estimate telephone connection performances.

19. Claims 29-31 are rejected under 35 U.S.C. 103(a) as being unpatentable over Farris'216, Randic'797 and ITU P.830, as applied to claims 26 above, and further in view of well established teaching in art and Sand'746.

Regarding claims 29-31, the combined system of Farris'216, Randic'797 and ITU P.830 teaches wherein said first communication service and second communication services as descried above in claim 26.

Neither Farris'216, Randic'797 nor ITU P.830 explicitly discloses said second communication service is subject to at least one impairment that does not affect the first communication service.

However, the above-mentioned claimed limitations are taught by well-established teaching in art. Well-established teaching in art teaches said second communication service is subject to at least one impairment that does not affect the first communication service. Farris'216 discloses the PSTN services in PSTN network as first communication services, and Internet services in Internet as second communication services. Randic'797 teaches measuring the quality parameters and defects/errors in the Internet. It is well known in the art that the defects, which

involve packets (i.e. packet loss or delay), are not measured in PSTN circuit switched services. Thus, the Internet service is subjected to at least one packet defect that does not affect the PSTN services.

In view of this, having the combined system of Farris'216, Randic'797 and ITU P.830, then given the teaching of well established teaching in art, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216, Randic'797 and ITU P.830, for the purpose of providing the fact that the Internet service is subjected to at least one packet defect that does not affect the PSTN services, as taught by well established teaching in art. The motivation being that determining unrelated defects while measuring the services offered in PSTN and Internet, it can improve the measurement accuracy by considering the related defect with applicable network.

Neither Farris'216 nor Randic'797 explicitly discloses packet loss (see Sand'746 FIG. 4, network performance packet loss 4 and voice performance echo path loss 4) and packet delay (see Sand'746 FIG. 4, network performance packet delay 5 and voice performance echo path delay 3) see col. 6, lines 20-35.

However, the above-mentioned claimed limitations are taught by Sand'746. In view of this, having the system of Farris'216, Randic'797 and ITU P.830, then given the teaching of Sand'746, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216, Randic'797 and ITU P.830, for the purpose of providing testing/measuring mechanism of the voice quality such as packet loss and delay

over the Internet and determination a result in order to improve the quality of voice path to conform to the SLA and ANSI level, as taught by Sand'746, since Sand'746 states the advantages/benefits at see col. 1, lines 46 col. 3, lines 67 that it would provide meaningful and accurate measurement of voice GOS which can impact the performance. The motivation being that by measuring and testing the voice communication paths over the Internet and utilizes the results to improve the quality of voice path, it can increase capability of the service provider to proactively test and measure the voice quality of the traffic sent over a packet network before the signal is totally degraded, and the measuring process can be provided through software.

20. Claims 27 and 28 are rejected under 35 U.S.C. 103(a) as being unpatentable over Farris (U.S. 6,574,216) and Randic (U.S. 6,275,797), in view of ITU P.830.

Regarding claim 27, the combined system of Farris'216, Randic'797 and well established teaching in art first, second quality indices and the percentage of total acceptable quality (i.e. P values) as described above in claim 26. Farris'216 further discloses determining that the second communication is unacceptable quality if threshed exceeds a threshold value as described above in claim 26.

Neither Farris'216 nor Randic'797 explicitly discloses wherein quality indices are P(UDI) values (see ITU P.830, see page 14-15, section 10.2, 10.2.1, listening quality scales or rating: wherein <u>U</u>nusable is bad, <u>D</u>ifficult is poor, and <u>I</u>rritating is fair), and P(UDI) value related to average portion of communication that would be rated as objectionable by users (see ITU P.830, see page 14-15, section 10.2, opinion scales and tables utilizes the opinion of users (i.e. listeners and talkers)

regarding the quality of speech as bad, poor, and fair scale, and the poor, fair and bad qualities are not acceptable to the user and they can be rated as objectionable by users; see page 6, section 8.1.3 talkers and page 10, sections 10.1 and 11. Also, when rating/scaling the speech quality among the listeners and talkers, the average data must be determined in order to collect quality data over multiple calls among users).

However, the above-mentioned claimed limitations are taught by ITU P.830. In view of this, having the combined system of Farris'216 and Randic'797, then given the teaching of ITU P.830, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the combined system of Farris'216 and Randic'797, for the purpose of providing the average score rating of poor, fair and bad opinions, as taught by ITU P.830, since ITU P.830 states the advantages/benefits at see page ii, introduction, paragraph 2 that it would ensure the performance of the complete system satisfactory. The motivation being that by testing and measuring the degradation contributed by the non-linear part of the transmission path, it can increase the reliability of the network and ability to estimate telephone connection performances.

Regarding claim 28, the combined system of Farris'216, Randic'797 and ITU P.830, teaches P(UDI) of the second communication service as described above in claim 26 and 27.

Neither Farris'216 nor Randic'797 explicitly discloses threshold value 0.06. Randic'797 teaches acceptable signal threshold of 0.75 or 75% (which means

acceptable error or defect threshold is 0.25 or 25%) provided by the ISP according to the quality of acceptable voice over the Internet. ITU P.830 teaches determining and performing various test with regards to quality of voice. Setting threshold to 0.06% does not define a patentable distinct invention over that in the combined system of Farris'216, Randic'797 and ITU P.830 since both the invention as a whole and the combined system of Farris'216, Randic'797 and ITU P.830 are directed to determining the threshold required for sending voice-o-IP traffic so as to maintain the voice quality. The degree in which determining threshold value presents no new or unexpected results, so long as the voice quality is maintained, the voice traffic is processed in a successful way. If one has less number of error thresholds to determine quality, it will be provide excellent or good service, and if one has more number of error thresholds, it will provide fair service. Therefore, to have threshold value of 0.06 that maintain quality of voice would have been routine experimentation and optimization in the absence of criticality.

Allowable Subject Matter

21. Claim 32 is allowed.

Response to Arguments

22. Applicant's arguments filed 11-17-2004 have been fully considered but they are not persuasive.

Regarding claim 1, the applicant argued that, "...Farris does not disclose or suggest obtaining a first quality index pertaining to the first communication service...second quality index pertaining to second communication services" in page 14, paragraph 2; page 15, paragraph 1.

Regarding claims 1-31, in response to applicant's argument, the **examiner respectfully disagrees** that Farris does not disclose or suggest obtaining a first quality index pertaining to the first communication service.

Farris discloses obtaining a first quality index (**see col. 4, lines 63-67; a voice quality threshold/criteria of normal end-to-end voice circuit which a caller accepted**) pertaining to the first communications service (**see FIG. 3, PSTN 10, note the performance quality acceptable to the calling subscriber is determined based upon circuit switched performance quality threshold/criteria utilized in PSTN, thus, it is relevant to PSTN services; see col. 8, lines 15-40**);

obtaining a second quality index (**see col. 10, lines 1-24, 44-65; a measured threshold/criteria of the packet switched network; see col. 10, lines 1-24, 44-65**), pertaining to the second communications service (**see FIG. 3, Internet 50, note the performance quality threshold/criteria is determined in Internet, thus it is relevant to Internet services; see col. 10, lines 1-24**); and

determining that the second communication service (**see FIG. 3, routing the traffic over Internet 50**) is of unacceptable quality if the second quality index differs from the first quality index service by more than a selected amount (**see FIG. 6A, step 208 and 214, quality good? N; see col. 10, lines 1-2, 44 to col. 11, lines 15;**

note that the routing the call over the Internet is unacceptable when measured/tested quality threshold does not meet the PSTN acceptable quality threshold by a predetermined amount. Note that in order to perform a comparison and selecting one service over the other, one must identify whether the difference are more than a predetermined amount).

Note that Farris discloses obtaining/determining the voice quality relevant or acceptable to PSTN services or PSTN users, so that the system can compare with the obtained/determined quality relevant to Internet services, so that the system can determine whether the voice quality relevant to the Internet is acceptable to use it. One cannot determine whether the quality is acceptable or unacceptable without the comparing to the other. In order to compare, one must first determine the acceptable quality index, as stated in Farris FIG. 3. Thus, Farris clearly anticipated the applicant claims invention.

Regarding claim 1, the applicant argued that, "...**PSTN...voice circuit...threshold**...this does not discloses ... in claim 1...." in page 15, paragraph 15.

Regarding claims 1-31, in response to applicant's argument that the references fail to show certain features of applicant's invention, it is noted that the applicant arguments are based upon the detailed information or features recited in Farris, yet none of this detailed or features information are recited in the rejected claim(s). Although the claims are interpreted in light of the specification, limitations

from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993).

Regarding claim 6, the applicant argued that, "...the combination of Farris and Randic fails to discloses...determining a value for the performance characteristic required to maintain the second quality index acceptable near the value of the first quality index ...." in page 17, paragraph 1-2.

Regarding claims 1-31, in response to applicant's argument, the **examiner respectfully disagrees** that the combined system of Farris and Randic fails to disclose above argued limitations.

Farris discloses obtaining a first quality index (**see col. 4, lines 63-67; a voice quality threshold/criteria of normal end-to-end voice circuit which a caller accepted**) pertaining to the first communications service (**see FIG. 3, PSTN 10, note the performance quality acceptable to the calling subscriber is determined based upon circuit switched performance quality threshold/criteria utilized in PSTN, thus, it is relevant to PSTN services; see col. 8, lines 15-40**);

determining the effect of at least one performance characteristic of the second communication service (**see col. 10, lines 15-24, 45-60; the quality performance measures/characteristic such as flow rate and response time which has the effect on the voice-o-IP network are determined**) upon a second quality index (**see col. 10, lines 1-24, 44-65; a measured threshold/criteria of the packet switched network; see col. 10, lines 1-24, 44-65**), pertaining to the second communications service (**see FIG. 3, Internet 50, note the performance quality**

threshold/criteria is determined in Internet, thus it is relevant to Internet services; see col. 10, lines 1-24); and

determining that the second communication service (**see FIG. 3, routing the traffic over Internet 50**) is of unacceptable quality if the second quality index differs from the first quality index service by more than a selected amount (**see FIG. 6A, step 208 and 214, quality good? N; see col. 10, lines 1-2, 44 to col. 11, lines 15; note that the routing the call over the Internet is unacceptable when measured/tested quality threshold does not meet the PSTN acceptable quality threshold** by a predetermined amount. Note that in order to perform a comparison and selecting one service over the other, one must identify whether the difference are more than a predetermined amount).

Randic discloses teaches obtaining a first quality index (**see FIG. 1, voice test file 23, which is the non-distorted common reference voice pattern, also see FIG. 3 steps 40 and 47; see col. 3, lines 15-39, see col. 6, lines 1-9**),

obtaining a second quality index (**see FIG. 1, transmitted voice test file 17, which is the distorted reference voice pattern, see FIG. 3, step 44)** pertaining to the second communions service (**see FIG. 1, WAN 11 or see FIG. 3, packet base network 42; test file 17 is routed through the packet switched network, thus it pertains/relevant to the packet switched network services; see col. 3, lines 40-60, see col. 5, lines 24-67**; and

determining a value (**see FIG. 3, step 52, determining voice quality factor 52, and voice quality factor 27; also see FIG. 1**), required to maintain the second

quality index acceptable near the value of the first quality index **(see col. 7, lines 1-29; note that a voice quality factor 27 is determine to improve and conform the reference acceptable threshold level of voice path defined by voice test file 23 (i.e. 75% match to the reference)**).

Thus, it is clear the combined system Farris and Randic disclosed the applicant claimed invention.

Regarding claim 6, the applicant argued that, "...the distorted and non-distorted data paths though network 42 are not analogous to the first and second communication services as recited in the claim ...." in page 18, paragraph 2.

Regarding claims 1-31, in response to applicant's arguments, the first and second communication services are disclosed in Farris as stated above. Randic discloses second communication service, WAN 11 (FIG. 1) or packet network 42 (see FIG. 3).

Regarding claim 4 and 5, the applicant argued that, "...Randic do not disclose computing an expected quality index for the second communication service...." in page 19, paragraph 2.

In response to applicant's argument, **the examiner respectfully disagrees** that the combined system of Farris and Randic fails to disclose above argued limitations.

Farris discloses measuring performance characteristics of the second network and computing an quality index (**see FIG. 6A, step 204; 206; during quality check and monitor step, a measured performance quality level (i.e.**

measured rate or measured response time) is acquired by the test signal from quality test application 122, see FIG. 5, col. 10, lines 1-24, 44-65. Also, **see FIG. 3, Internet 50, note the performance quality threshold/criteria is determined in Internet; see col. 10, lines 1-24**).

Randic discloses an expected quality index (**see** FIG. **3, step 52, determining voice quality factor 52, and voice quality factor 27; also see FIG. 1**) for the second communications service (**see FIG. 3, Internet 50 service); see col. 7, lines 1-29.)**

In response to applicant's arguments against the references individually, one cannot show nonobviousness by attacking references individually where the rejections are based on combinations of references. See *In re Keller*, 642 F.2d 413, 208 USPQ 871 (CCPA 1981); *In re Merck & Co.,* 800 F.2d 1091, 231 USPQ 375 (Fed. Cir. 1986). Thus, the combined system of Farris and Randic clearly discloses the claimed limitations, and the combination is proper.

Regarding claim 4-5, the applicant argued that, "...**voice path...voice quality factor...**is not an expected quality index, as required by claim 4...." in page 19, paragraph 3.

Regarding claims 1-31, in response to applicant's argument that the references fail to show certain features of applicant's invention, it is noted that the applicant arguments are based upon the detailed information or features recited in Randic, yet <u>none</u> of this detailed or features information are recited in the rejected claim(s). Although the claims are interpreted in light of the specification, limitations

from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993).

Regarding claim 22, the applicant argued that, "...Sand, Shaffer, and well established teaching in art fails to disclose or suggest comporting an alerted data set by changing, in the original data set, the occurrences of values of the second performance characteristic assuming the first performance characteristic is set to said given value..." in page 22, paragraph 4; page 23, paragraph 1; page 25, paragraph 1.

Regarding claim 22, in response to applicant's argument, the examiner respectfully disagrees that the combined system of Sand, Shaffer and well-established teaching in art fails to disclose above argued limitations.

Sand discloses assuming the first performance characteristic is set to said given value (see **FIG. 4, steps 44, 52 and 54; see FIG. 4, note that the standard delay values defined in ANSI T.221 and ANSI T1 LB 566 GOS standard are determined according to the acceptable QoS in industry; see col. 6, lines 20-35). Shaffer** disclose computing an altered data set by changing (**see FIG. 3, step 80; present QoS Table is updated**), in the original data set (**see FIG. 2, Present time QoS table**), the occurrences of values of the second performance characteristic (**see col. 6, lines 5-44; delay, latency, loss; note that based upon delay arrival or lost packet is assessed, the present time QoS table is updated**).

Thus, the combined system of Sand, Shaffer and well established teaching in art clearly discloses the claimed limitations.

Regarding claim 22, the applicant argued that, "…**QoS table…lost packets…delay to one another…**Shaffer does not disclose…." in page 24, paragraph 1.

Regarding claims 1-31, in response to applicant's argument that the references fail to show certain features of applicant's invention, it is noted that the applicant arguments are based upon the detailed information or features recited in Sand and Shaffer, yet <u>none</u> of this detailed or features information are recited in the rejected claim(s). Although the claims are interpreted in light of the specification, limitations from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993).

In view of the above, **the examiner respectfully disagrees** with applicant's argument and believes that the references as set forth in the 102 and 103 rejections are proper.

Conclusion

23. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not

mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

24. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Ian N Moore whose telephone number is 571-272-3085. The examiner can normally be reached on M-F: 9:00 AM - 6:00 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Chau T Nguyen can be reached on 571-272-3126. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

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